



Business Communication Manager Release 5.0 Configuration Guide

For Use with AT&T IP Flexible Reach, IP Flexible Reach with Business in a Box,
and IP Toll Free

Issue 2.1

Abstract

This document provides guidelines for configuring a SIP Trunk between a BCM Release 5.0 and AT&T IP Flexible Reach, IP Flexible Reach with Business in a Box and IP Toll Free.

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Introduction

This document provides a configuration guide to assist Avaya BCM Release 5.0 administrators in connecting to AT&T IP Flexible Reach, IP Flexible Reach with Business in a Box, and IP Toll-Free via SIP trunks.

This document does not describe procedures to configure the BCM for advanced functionality. For more information and procedures, please refer to the Avaya technical documentation found on the Avaya website.

Pre-IP PBX Configuration Activity

This guide assumes that the administrator is knowledgeable in IP PBX programming and operations.

An important tool that the administrators should have at their disposal prior to testing their IP PBX with AT&T IP Flexible Reach is the Wireshark network protocol analyzer. This software can be used to run and analyze traces on problem calls so the information can be shared with equipment and network engineers. This free software can be obtained at <http://www.wireshark.org/>.

The customer may also use TCPDUMP which can be found on most UNIX and Linux systems. The customer should have Wireshark or TCPDUMP loaded on a server that is connected to a LAN switch or hub that can monitor both the signaling and media packets on any calls between the customer PBX and the IP Flexible Reach managed router.

BCM Release 5.0 introduces a built-in packet capture tool that can be used to capture SIP signaling packets originating from and going to the BCM. These files can be downloaded from the BCM to a computer and viewed. To view these files, Wireshark (or similar) protocol analyzer is required on the destination computer.

Customer Questions

Section 4 of this guide provides screen shots and instructions for the configuration of your IP PBX. Should you have questions regarding these instructions, please contact Brian Stegemoller at +1 (972) 745-5139. When calling this number please have the following information available:

- Company name
- Company location
- Administrator name and phone number
- IP PBX name and software version
- Customer Configuration Guide – Issue number and date

Trouble Reporting

Avaya and AT&T will make every effort to quickly resolve reported troubles. The time required for trouble shooting can be reduced if the customer has the necessary detailed information available when reporting a problem. Prior to reporting a problem please provide a Wireshark or TCPDUMP trace of the failed call.

Document Feedback

IP PBX administrators who would like to provide feedback on the contents of this document should send it to Brian Stegemoller (brianstegemo@avaya.com) with a copy to Albert Chee (alchee@avaya.com) and Steven Chen (stevenchen@avaya.com).

Document Change History

Issue 0.0	November 8, 2009; Draft
Issue 1.0	November 18, 2009; Included "50/450" in header and title; Added entries in Special Notes; Updated patch information
Issue 1.1	December 15, 2009; Added IP Toll-Free support
Issue 2.0	May 26, 2010; Added IP Flexible Reach with Business in a Box support
Issue 2.1	July 27, 2010; Changed document template from Nortel to Avaya

Special Notes

Emergency 911/E911 Services Limitations

While AT&T IP Flexible Reach services support E911/911 calling capabilities in certain circumstances, there are significant limitations on how these capabilities are delivered. Please review the "AT&T IP Flexible Reach" Service Guide in detail to understand these limitations and restrictions.

BCM Soft Phones not currently supported

Avaya BCM soft phones are not currently supported with the AT&T IP Flexible Reach Services.

Ring back Issues with Unattended Transfers

An unattended transfer is one in which the party initiating the transfer hangs up prior to answer by the party to whom the call is being transferred. When 2 phones are in an active call on the BCM and one of those phones performs an unattended transfer to certain endpoints on the AT&T network, the BCM phone remaining on the call will not hear ring back prior to answer.

T.38 Fax must be used

T.38 fax should be configured on the BCM with AT&T IP Flexible Reach service. Fax via G.711 transport should not be used on the BCM.

Error Correction Mode Not Supported

In T.38 fax calls originated by the BCM, during capabilities negotiation, BCM disables Error Correction Mode (ECM); therefore, these particular calls will not use ECM during fax transmission. However, calls will still work.

Find Me/Follow Me Restrictions

Directory Information (i.e., 411 and 555-1212) and 8YY numbers should not be used for Find Me/Follow Me external destinations.

20-byte G.729 Payload Size to/from PSTN

The G.729 payload size will always be 20 bytes with calls to/from PSTN.

G.726 Codec not supported

BCM does not support the G.726 codec – G.711 and G.729 codecs should be used.

Alternate Destination Routing (ADR) with Intelligent Contact Center (ICC)

ADR RNA will be invoked on inbound IP Toll Free calls to BCM ICC only when the destination skillset is disabled (not in operation). In all other cases, calls will be considered answered.

ADR on BUSY will not be invoked on inbound IP Toll Free calls to BCM ICC since calls will be queued.

Hold and Resume Call Failures with TDM PBX behind AT&T Managed Router

There is a known issue with calls to an IP Flexible Reach customer running a TDM PBX behind an AT&T managed router. When a BCM user places this call on hold, the call fails. Equipment upgrades are planned in the AT&T network in late 2010 to fix this issue.

Overview

This section provides a service overview of the Avaya Business Communication Manager (BCM) Release 5.0 IP PBX integration with AT&T IP Flexible Reach services.

Release 5.0 introduces a common software release for both BCM50 and BCM450. Thus, "BCM" will refer to BCM Release 5.0 and will not refer to any particular hardware platform (BCM50 or BCM450).

- Customer Premises
 - Phones and IP PBX Server in Private Address space
 - AT&T Managed Router
- Customer Sites connect to ATT IP Border Element (IPBE)

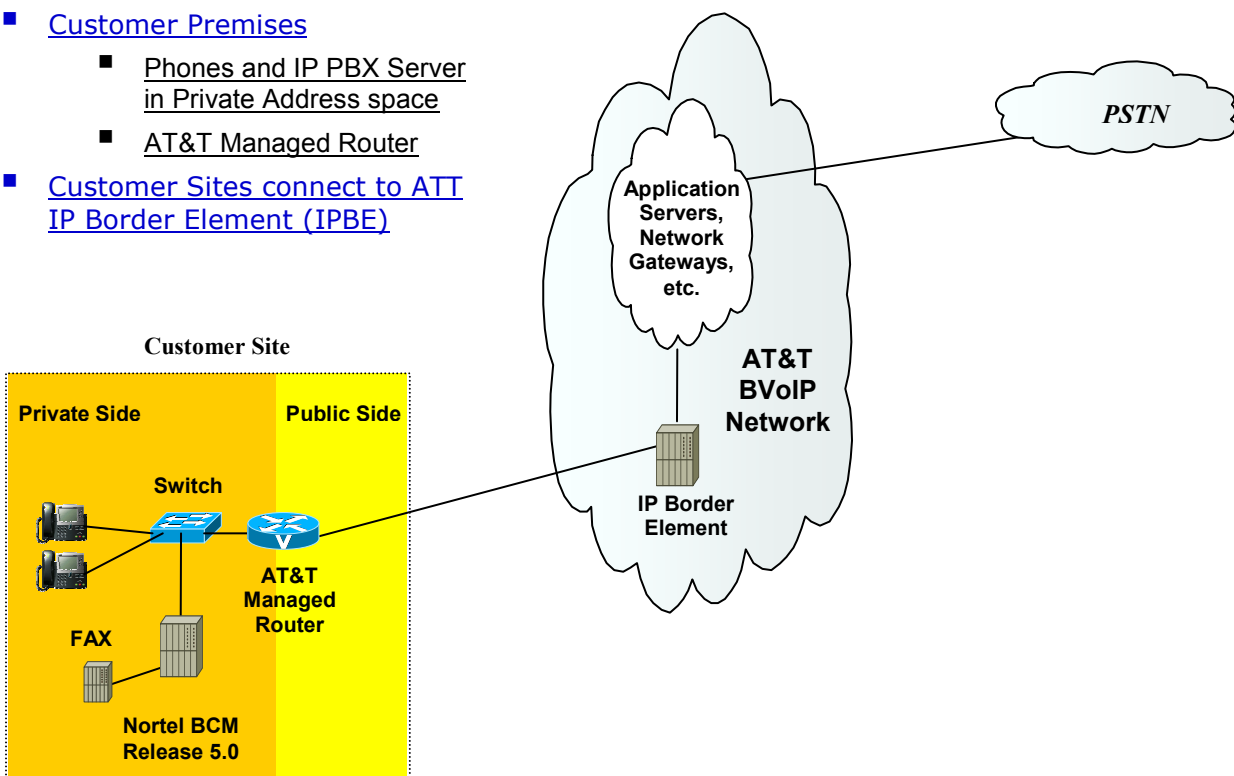


Figure 1: AT&T BVoIP Network

The Avaya BCM customer premises site shall consist of the following components.

- Avaya IP 200x, 11xx, 12xx phones* – These phones use the Avaya proprietary UNISTim signaling protocol to communicate to the Avaya BCM IP PBX for call feature and routing support. These phones can be connected to a Avaya Ethernet switch (ES 470, ERS 5520, etc.) that supplies in-line power (IEEE 802.3af) to the phones.

- The following interfaces and applications can also be used on the BCM. They include (but are not limited to):
 - Digital station ports supporting digital phones
 - Analog station interfaces with message waiting and CLID support
 - Integrated CallPilot voice mail system, auto attendant, and meet-me conferencing
 - Analog trunk module for analog/POTS lines to PSTN
 - Digital trunk module for PRI/T1 connection to PSTN

* RFC2833 DTMF is currently not supported on the IP Softphone 2050; thus the IP Softphone 2050 is not supported for AT&T IP Flexible Reach services.

The following routing scenarios are supported by the Avaya BCM IP PBX and **DO NOT** use the AT&T Call Control.

- Local Avaya BCM phone to other local Avaya BCM phones

The following routing scenarios are supported by the Avaya BCM IP PBX and **DO** use the AT&T Call Control. For voice calls, the G.729 codec shall be used.

- Avaya BCM phones to PSTN (domestic US and international).
- Avaya BCM phones to legacy PBX site with Cisco gateway.
- Legacy PBX site with Cisco gateway to Avaya BCM phones.
- Avaya BCM phones at one Avaya BCM IP PBX site to Avaya BCM phones at another Avaya BCM IP PBX site

If the customer has subscribed to Calling Plans B and C (Local), then the following routing scenarios are supported by the BCM IP PBX and **DO** use the AT&T Call Control. For voice calls, the G.729 or G.711 codec may be used.

- Inbound PSTN to BCM phone
- Outbound local PSTN calls from the BCM phones
- Outbound local N11 (i.e. 411, 911) calls from the BCM phones

T.38 Fax was tested and is supported on the BCM with the AT&T IP Flexible Reach services to/from the following:

- PSTN
- Legacy PBX site with Cisco gateway
- Another BCM IP PBX site

BCM Release 5 and IP Flexible Reach

This configuration guide specifies the Avaya BCM screens that must be configured and updated to support the AT&T IP Flexible Reach services.

Avaya BCM Version and Feature Requirements

The Avaya BCM must be running at least software version 9.0.1.22.XXX. You can check the version of BCM by viewing the following screen under **Administration → Software Management → Software Update History**.

This is the supported base release that is required for AT&T IP Flexible Reach services.

This software release 9.0.1.22.XXX refers to BCM Release 5.0.

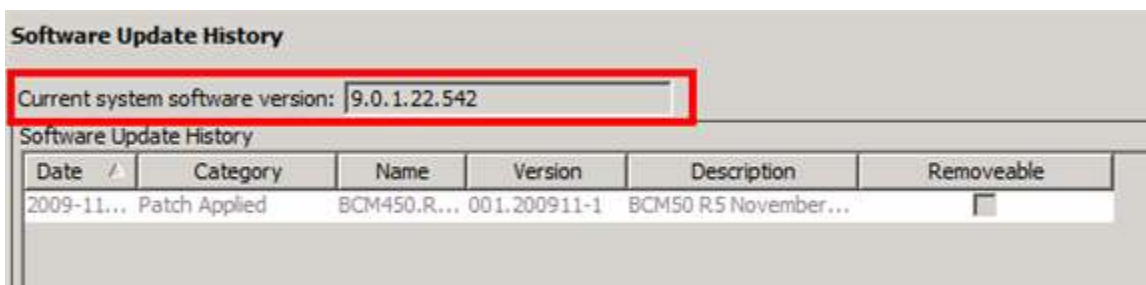


Figure 2: BCM Software Version Number and Patch List

The following BCM patches must be applied. To verify any installed patches on the system: **Administration → Software Management → Software Update History**.

Patch Name/ID	Description
BCM050.R500.SU.System-002.200912 (or greater)*	BCM50 RIs 5.0 December Smart Update To resolve a specific call forward scenario, where PSTN gateway sends 183 Session Progress and 180 Ringing SIP messages, which results in no media heard.
BCM450.R500.SU.System-002.200912 (or greater)*	BCM450 RIs 5.0 December Smart Update To resolve a specific call forward scenario, where PSTN gateway sends 183 Session Progress and 180 Ringing SIP messages, which results in no media heard.

*Note: Avaya recommends installing the latest SU, which will contain previous fixes and new fixes.

IP Trunks

Voice over IP (VoIP) trunks, are signaling channels that simulate how CO lines work. However, VoIP trunks transmit data to the IP network over a LAN or IP network rather than over physical lines. Once the VoIP trunks are set up, you can assign them to line pools, and program their behavior in the same way you would PRI lines.

VoIP trunks use line numbers, which appear under ***Configuration → Telephony → Lines → Active VoIP Lines***. To access VoIP lines, you need to enter software keycodes. Each keycode supports a specific number of trunks. No entries appear in the Enabled VoIP lines field until you complete the IP Trunks Settings field, which displays when you click IP Trunks under ***Configuration → Resources → Telephony Resources → IP trunks***.

Note: The BCM offers two VoIP trunk license options: SIP Gateway Trunk License and VoIP Trunk Gateway License. The SIP Gateway Trunk License enables SIP-only trunks and the VoIP Trunk Gateway License enables SIP or H.323 trunks. Either type of trunk licenses can be used for SIP signaling with AT&T IP Flexible Reach service.

Customers that desire a lower cost or have no requirements for H.323 should choose the SIP Gateway Trunk License option.

VoIP trunks should be configured to use a single line pool. Do not mix other trunk types on the same line pool (e.g. analog, PRI, etc). The VoIP line pools are assigned to routes, which, in turn, are configured with destination codes that route calls to the BVoIP network.

Check under ***Configuration → Telephony → Lines → Active VoIP Lines*** to see if trunks have been allocated. You should have a number of IP trunks displayed. The total number of lines indicated corresponds to the number of IP trunks licensed by Avaya for your BCM. See figure below.

Line	Trunk Type	Name	Control Set	Line Type	Prime Set	Pub. Received #
001	VoIP	Line001	5000	Pool:BlocA	5000	N/A
002	VoIP	Line002	5000	Pool:BlocA	5000	N/A
003	VoIP	Line003	5000	Pool:BlocA	5000	N/A
004	VoIP	Line004	5000	Pool:BlocA	5000	N/A
005	VoIP	Line005	5000	Pool:BlocA	5000	N/A
006	VoIP	Line006	5000	Pool:BlocA	5000	N/A
007	VoIP	Line007	5000	Pool:BlocA	5000	N/A
008	VoIP	Line008	5000	Pool:BlocA	5000	N/A

Figure 3: Available VoIP Trunks

Note: If no active VoIP lines are present, check to see if either SIP or IP Trunks licenses are installed. If so, try restarting the “feps” service on the BCM. This can be done by going to **Administration → General → Service Manager**, and restarting the “feps” service.

Also, under **Configuration → Resources → Application Resources**, ensure that the SIP and/or IP Trunks Minimum and Maximum values are set to 0 and MAX, respectively.

Application Resources

Total Resources: Signalling channels: 580, VDI channels: 194, Media channels: 1410, DSP resources: 180

Reserved Resources: Signalling channels: 8, VDI channels: 0, Media channels: 10, DSP resources: 10

Application	Minimum	Maximum	Licence	System Max.	Change Pending
CTE Terminals	0	MAX	N/A	32	
Conf. Mixers	0	30	N/A	62	
Conf. Parties	0	61	N/A	124	
Fax	0	MAX	0	8	
IP Sets	0	MAX	4	480	
IP Trunks	0	MAX	0	192	
IVR	2	MAX	0	24	
Media Gateways	2	MAX	N/A	672	
Nortel SIP Sets	0	MAX	0	480	
Other SIP Sets	0	MAX	0	480	
SIP Trunks	0	MAX	8	192	
Voice Mail + CC	8	48	N/A	63	

Figure 4: Application Resource availability of SIP (or IP) Trunks

Also, ensure that there is a number under “**Licence**” for either trunk entries. Otherwise, new keycodes/licenses need to be applied to the BCM.

Configuration → Telephony → Dialing Plan → Line Pools:

In this case we selected "BlocA" under the "Pool" column. We will use this line pool to access the VoIP trunks. Additionally, all DN numbers that need to access the VoIP trunks must be added to this pool. In other words, ensure that the DN numbers have the same line pool assigned as the VoIP trunks (in Figure 4). Please see the following screen shots for example configurations.

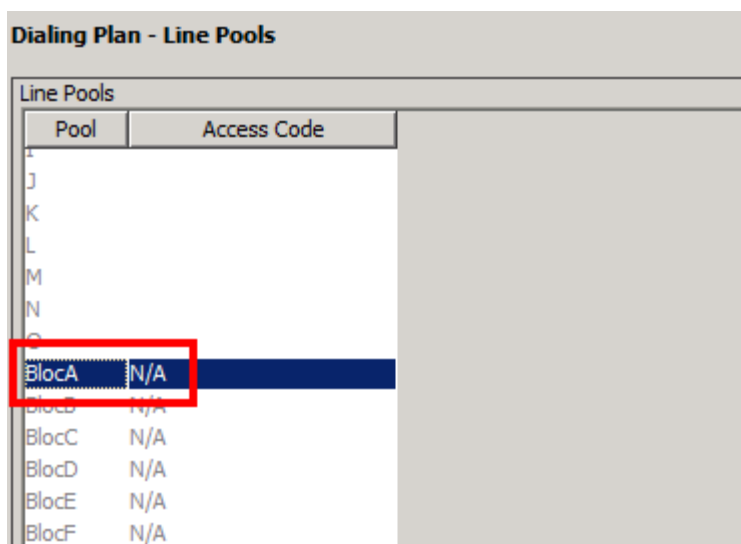


Figure 5: Assigning Line Pool to IP Trunks

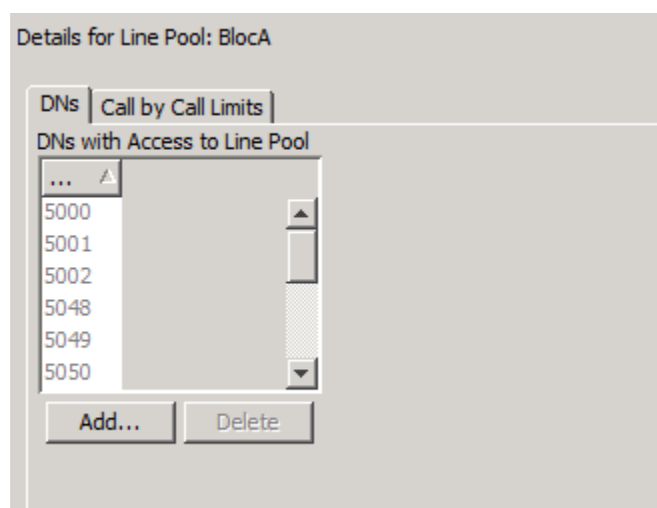


Figure 6: Assigning DN to Line Pool

Configuration → Telephony → Dialing Plan → Public Network:

We define the Public Received number length to "4" digits and Public network dialing plan to "National." This tells the BCM to look at the last four digits of a public received number, and alert the assigned DN of an incoming call.

DN Prefix	DN Length
0	11
00	12
01	17
1	11
011	18
411	3
911	3
Default	7

Code Prefix	ID Length
10	3
101	4

Figure 7: Configuring Public Received Number Length and Dialing Plan

Configuration → Telephony → Dialing Plan → Routing:

Select the "**Routes**" tab and ensure there is an entry for "BlocA" with DN Type "National." In this case "100" is the route number and the DN Type is specified as "National." See figure below.

Route	External Number	Use Pool	DN Type	Service Type	Service ID
000		A	N/A	N/A	N/A
100		BlocA	National	N/A	N/A

Figure 8: Assigning a Route for IP Trunks

Configuration → Telephony → Dialing Plan → Routing:

Select the “**Destination Codes**” tab to assign a destination code for the IP trunks. Configure a destination code (i.e., “9”) for access to outside destinations that will be presented to the AT&T service for routing. In this case, when “9” is dialed we wish to push the dialed string to the IP trunk for routing. See figure below.

“Absorbed Length” is configured for “All” so that the destination code string will not be sent to AT&T with the number. “Normal Route” is set to “100”, which was configured in the previous step for the IP trunks.

Note: When completing the Technical Questionnaire Section 6.0 Dial Plan Information and the **private dial plan** is selected (**YES**), then please refer to *Appendix 'A': Configuring Destination Code with Wildcard*.

Dialing Plan - Routing

Routes | **Destination Codes** | Second Dial Tone

Destination Code	Normal Route	Absorbed Length	Wild Card: 0	1	2	3	4	5	6	7	8	9
9	100	All										

Add... Delete

Alternate Routes for Destination Code: 9

Schedule	First Route	Absorbed Length	Second Route	Absorbed Length	Third Route	Absorbed Length
Evening	All		All		All	
Lunch	All		All		All	
Night	All		All		All	
Sched 4	All		All		All	
Sched 5	All		All		All	
Sched 6	All		All		All	

Figure 9: Assign Destination Code to Access IP Trunk Routes

IP Trunk Settings

Configuration → Resources → Telephony Resources:

Select module type “**IP Trunks**” and click on the “**IP Trunk Settings**”.

The screenshot shows the 'Telephony Resources' configuration window. At the top, there is a table listing various modules. The 'Internal IP Trunks' module is highlighted. Below the table, there are buttons for 'Disable', 'Enable', 'Deconfigure...', and 'Configure...'. The 'Details for Module: Internal IP Trunks' section is expanded, showing several tabs: 'Routing Table', 'IP Trunk Settings', 'H323 Settings', 'H323 Media Parameters', 'SIP Settings', 'Sip Proxy', and 'SIP Med'. The 'IP Trunk Settings' tab is selected, and within it, the 'Telephony Settings' section is visible. Two options are checked and highlighted with red boxes: 'Forward redirected OLI:' and 'Send name display:'. Other options like 'Remote capability MWI:' and 'Ignore in-band DTMF in RTP:' are also visible but not checked.

Location	Configured Device	Dip Switch	Bus	State	Low
Internal	IP Trunks	N/A	N/A	Enabled	001
Internal	IP Sets	N/A	N/A	Enabled	5000
Internal	Applications	N/A	N/A	Enabled	5100
Main MBM 1	None	N/A	N/A	N/A	N/A
Main MBM 2	None	N/A	N/A	N/A	N/A
Main MBM 3	CTM4/GATM4 MBM	All On	30.1	Enabled	009
Main MBM 4	ASM/ASM+ MBM	All On	40.1	Enabled	5048
Expansion 1	None	N/A	N/A	N/A	N/A

Buttons: Disable, Enable, Deconfigure..., Configure...

Details for Module: Internal IP Trunks

Tabs: Routing Table, IP Trunk Settings, H323 Settings, H323 Media Parameters, SIP Settings, Sip Proxy, SIP Med

Telephony Settings

Forward redirected OLI: ☒ Send name display: ☒

Remote capability MWI: ☒ Ignore in-band DTMF in RTP: ☐

Figure 10: IP Trunk Module and IP Trunk Settings

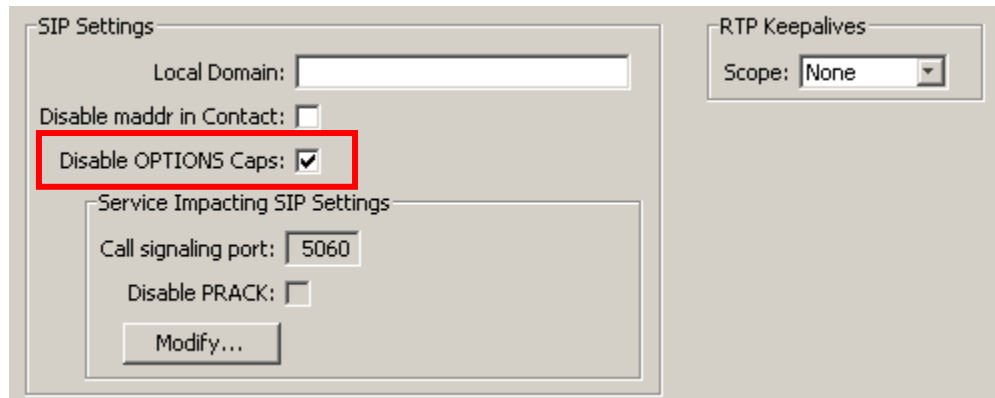
Ensure “Forward redirected OLI” and “Send name display” options are checked.

SIP Settings

Configuration → Resources → Telephony Resources:

Select module type “**IP Trunks**” and click on the “**SIP Settings**”.

Ensure that Disable OPTIONS Caps is **checked**.



The screenshot shows the 'SIP Settings' configuration window. It has a 'Local Domain' text field. Below it are two checkboxes: 'Disable maddr in Contact' (unchecked) and 'Disable OPTIONS Caps' (checked, highlighted with a red rectangle). Under the 'Service Impacting SIP Settings' section, there is a 'Call signaling port' field set to '5060' and a 'Disable PRACK' checkbox (unchecked). A 'Modify...' button is located at the bottom of this section. To the right, the 'RTP Keepalives' section shows a 'Scope' dropdown menu set to 'None'.

Figure 11: SIP Settings

SIP Proxy Parameters

Configuration → Resources → Telephony Resources:

Select module type “**IP Trunks**” and click on the “**Sip Proxy**” tab (see figures below).

The screenshot shows the 'Sip Proxy' configuration window. The 'SIP Proxy' tab is selected. The 'Domain' field is highlighted with a red box and contains the value '135.25.29.74'. The 'Route all calls using proxy' checkbox is unchecked. The 'MCDN Protocol' dropdown is set to 'None'. The 'Optional IP Address for legacy routing' section has an 'IP Address' field and a 'Port' field set to '0'. Below this is an 'Outbound Proxy Table' with columns: Name, IP Address, Port, Load-balancing Weight, and Keep alive. The table is currently empty. At the bottom of the table are 'Add...' and 'Delete' buttons.

Figure 12: Sip Proxy Parameters

For IP Flexible Reach: Populate the “Domain” as the IP address of the AT&T IP Border Element.

For IP Flexible Reach with Business in a Box: Populate the “Domain” as the IP address of the LAN interface of the AT&T Business in a Box router.

Contact your local AT&T Customer Care representative for the IP addresses and more information.

Failover to Secondary AT&T IP Border Element (IPBE)

This applies to AT&T IP Flexible Reach only. For AT&T IP Flexible Reach with AT&T Business in a Box customers, please skip this section.

Backup to a secondary AT&T IP Border Elements is supported on the BCM using the SIP OPTIONS keep-alive approach. The BCM will send SIP OPTIONS messages to the AT&T IPBEs listed in the Outbound Proxy Table and send VoIP calls based on whether or not the BCM receives a SIP response from the IPBEs and the Load-balancing Weight values set for each entry.

Media Parameters

Configuration → Resources → Telephony Resources:

Select module type “**IP Trunks**” and click on the “**SIP Media Parameters**” tab.

Within this screen; ensure that all values are exactly as the sample screen shot shown below:

- 1st Preferred Codec: G.729
- 2nd Preferred Codec: G.711-uLaw
- Voice Activity Detection: Disabled*
- Jitter Buffer: Auto
- Fax transport: T.38**
- G.729 payload size: 20
- G.711 payload size: 20

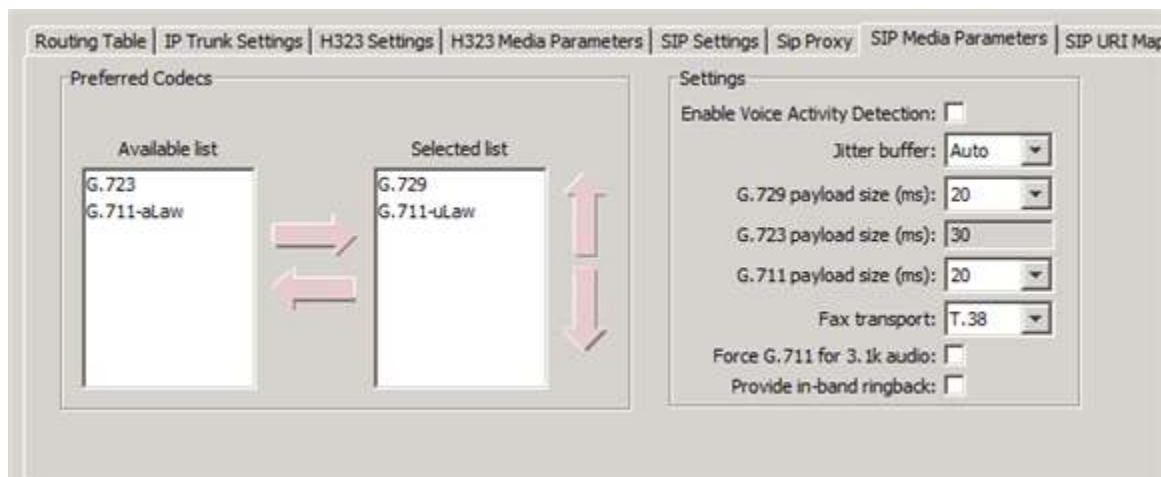


Figure 14: SIP Media Parameters

* For default configurations, G.729A codec will be used for voice calls. However, in the case that G.729B needs to be configured instead of G.729A, ensure that “Enable Voice Activity Detection” is checked.

** T.38 is the recommended fax transport.

SIP URI Map

Configuration → Resources → Telephony Resources:

Select module type “**IP Trunks**” and click on the “**SIP URI Map**” tab.

Ensure that the “e.164 / National” and “Unknown / Unknown” SIP domain names, at minimum, are blank.



The screenshot shows the 'SIP URI Map' configuration window. The 'SIP Domain Names' section contains the following entries:

Domain Name	Value
e.164 / National:	
e.164 / Subscriber:	subscriber.e164
e.164 / Unknown:	unknown.e164
e.164 / Special:	special.e164
Private / UDP:	udp
Private / CDP:	cdp
Private / Special:	special.private
Private / Unknown:	unknown.private
Private / Subscriber:	subscriber.private
Unknown / Unknown:	

The 'e.164 / National:' and 'Unknown / Unknown:' entries are highlighted with red boxes, indicating they should be blank.

Figure 15: SIP URI Map

Port Ranges

Configuration → Resources → Telephony Resources → Port Ranges:

The default RTP ranges are from 28000-28255 and 30000-30999. These ranges are used for fax, digital phones and analog phones and can be left as is. The media gateway port ranges are configurable.

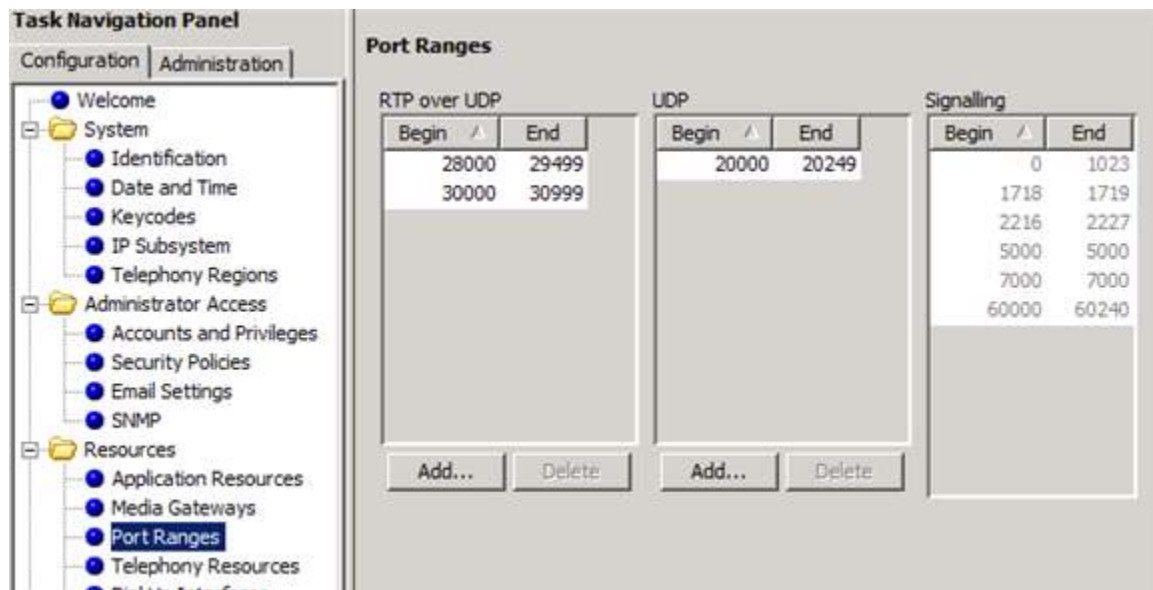


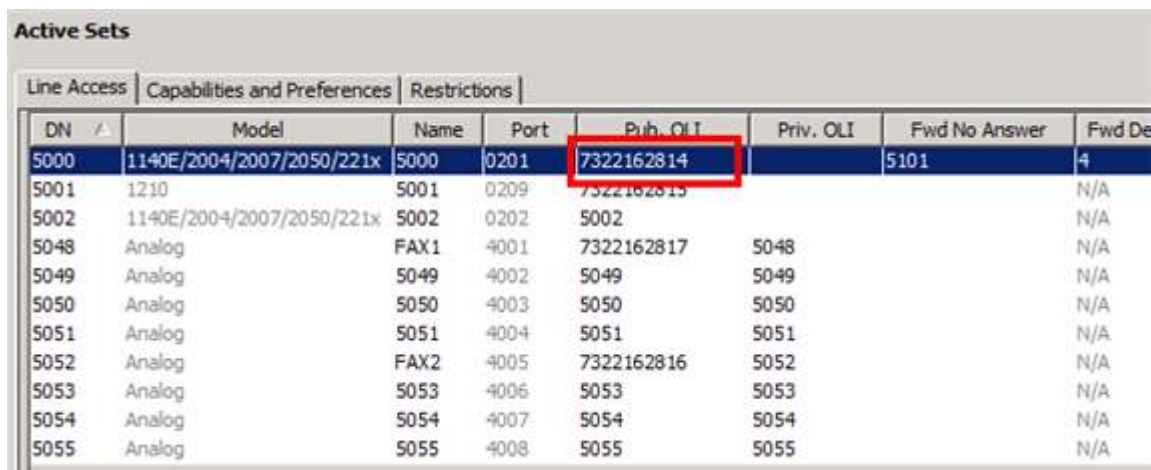
Figure 16: Media Gateway Port Ranges

The BCM IP phone's RTP and RTCP port ranges are 51000-51399. Each IP phone call uses two ports. The default port range for RTP and RTCP are not configurable.

Configuring Outgoing Calls from a BCM DN to AT&T IP Flex Reach

Configuration → Telephony → Sets → All DNs (or Active Sets):

We will now associate the DN extension with the DID number. In the example below, 7322162814 is entered in the "Public OLI" field. This example enables "calling number translation" (outgoing) for this particular DN number. See figure below.



DN	Model	Name	Port	Pub. OLI	Priv. OLI	Fwd No Answer	Fwd De
5000	1140E/2004/2007/2050/221x	5000	0201	7322162814		5101	4
5001	1210	5001	0209	7322162815			N/A
5002	1140E/2004/2007/2050/221x	5002	0202	5002			N/A
5048	Analog	FAX1	4001	7322162817	5048		N/A
5049	Analog	5049	4002	5049	5049		N/A
5050	Analog	5050	4003	5050	5050		N/A
5051	Analog	5051	4004	5051	5051		N/A
5052	Analog	FAX2	4005	7322162816	5052		N/A
5053	Analog	5053	4006	5053	5053		N/A
5054	Analog	5054	4007	5054	5054		N/A
5055	Analog	5055	4008	5055	5055		N/A

Figure 17: Configuring DID for Outgoing Calls

Configuring Incoming Calls from AT&T IP Flex Reach to a BCM DN

Configuration → Telephony → Sets → All DNs (or Active Sets):

We will now configure the “called number translation” (incoming) for the DN number. In our example, go to the “**Line Assignment**” tab located at the bottom of the “**Line Access**” page. Enter the last four digits (or however many digits specified in Figure 9) of the DID (Public number) in the “**Pub. Received #**” column.

Incoming DID calls will be routed to telephones, based on the trailing portion of the digits received by the network. For example, incoming calls from the AT&T IP Flexible Reach network will deliver a ten digit DID number, e.g. 7322162814. The BCM will process the called digit string using the last four digits, e.g. 2814, which is mapped to DN 5000. Additionally, this configuration will allow incoming 4-digit dialing plan calls from the IP Flexible Reach network, e.g. 2814.

Details for DN: 5000

Line Assignment Line Pool Access Answer DNs MeetMe Conferencing							
Assigned Lines							
Line	Appearance Type	Appearances	Caller ID Set	Vmsg Set	Priv. Received #	Pub. Received #	
361	Appr&Ring	1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	5000	2814	
362	Appr&Ring	1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>		3814	

Add... Delete

Figure 18: Configuring DID for Incoming Calls

Configuration → Telephony → Lines → Target Lines:

To display the DID number on the IP phone LCD screen, click on the assigned "Line" number of the DN you want to program. In our example below, we click on "**Line 361**"; enter "2162814" in the "**Name**" field. See figure below.

Target Lines						
Line ▲	Trunk Type	Name	Control Set	Line Type	Prime Set	Pub. Received #
361	Target line	2162814	5000	Public	5000	2814
362	Target line	3323814	5000	Public	5000	3814
363	Target line	Line363	5000	Public	5000	3815
364	Target line	Line364	5000	Public	5000	2815
365	Target line	2162816	5000	Public	5000	2816
366	Target line	2162817	5000	Public	5000	2817
367	Target line	Line367	5000	Public	5000	3817
368	Target line	3323816	5000	Public	5000	3816
369	Target line	Line369	5000	Public	5000	
370	Target line	Line370	5000	Public	5000	
371	Target line	Line371	5000	Public	5000	
372	Target line	Line372	5000	Public	5000	
373	Target line	Line373	5000	Public	5000	
374	Target line	Line374	5000	Public	5000	
<div>CopyPaste...Renumber</div>						

Figure 19: Display DID on IP Set LCD

Additionally, all telephone sets that need to access the VoIP trunks need to be configured with the designated "**Line Pool**" code. In our example, for DN 5000, we defined "**BlocA**" as the code to access the VOIP trunks. See figure 19 below.

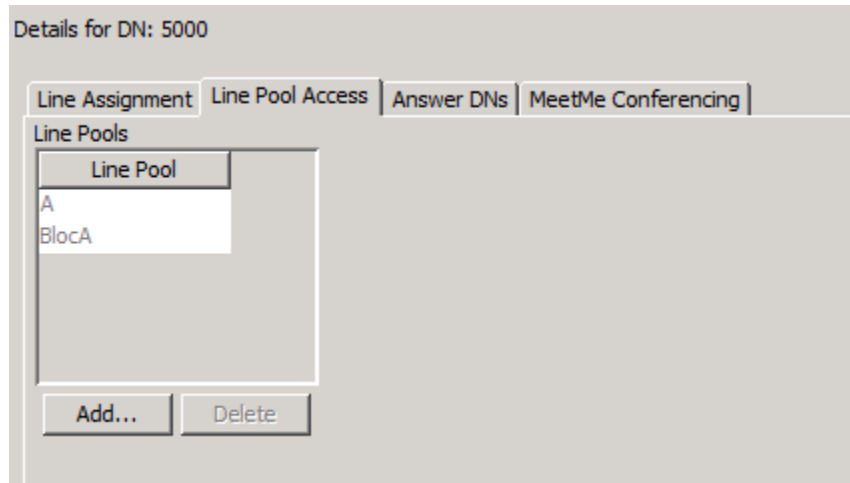


Figure 20: Assign Line Pool to IP Sets

BCM Release 5 and IP Toll-Free

BCM Release 5 was also tested with AT&T IP Toll-Free service. It can be used with the following setup (but not limited to):

- Directly to Phones
- Hunt Groups
- CallPilot – Auto-Attendant, Custom Call Routing (CCR) Trees, Intelligent Contact Center

Mapping DNIS to Target Lines

Configuration → Telephony → Lines → Target Lines:

Select the desired target line. Set the last x digits of DNIS (based on “**Public Received number length**” in Section 4.2) to the “**Pub. Received #.**” Set the “**Name**” of the target line to the DNIS digits (maximum is 7).

Target Lines							
Line	Trunk Type	Name	Control Set	Line Type	Prime Set	Pub. Received #	Priv. F
993	Target line	Line993	5000	Public	5000		
994	Target line	Line994	5000	Public	5000		
995	Target line	0001021		Public	5000	1021	
996	Target line	0001020		Public	5000	1020	
997	Target line	0001019		Public	5000	1019	
998	Target line	0001018		Public	5000	1018	
999	Target line	0001017		Public	5000	1017	

Figure 21: Target Line Configuration

Configuring Intelligent Contact Center

Contact Center is an application that handles incoming calls as efficiently and economically as possible. Contact Center answers calls, then routes the calls to agents in a skillset that most closely matches the needs of the caller. Calls can be routed based on the origin of the call, the destination of the call, or the information entered by the caller. Callers can be given high or low priorities. Callers can overflow to different groups or skillsets of agents, transfer out of the system, leave a message, and hear announcements or informative messages.

Note: Before setting up Contact Center, ensure that the appropriate keycode licenses are applied on the BCM.

Set up the Contact Center per customer requirements. This includes (but is not limited to):

- Agents
- Skillsets
- CLID/DNIS Routing Table (optional)
- Caller Input Rules (optional)

For more information on configuring Intelligent Contact Center, refer to the *Intelligent Contact Center Set Up and Operation Guide* (NN40170-301).

Configuring Auto-Attendant and CCR Trees

CallPilot has two powerful call routing features: Auto Attendant (AA) and Custom Call Routing (CCR). These features route incoming calls to telephones and voice mailboxes. You can also use these features to route calls to Contact Center skillsets.

Set up the Auto-Attendant and CCR Tree features. A sample is shown below:

CCR Tree Properties

Tree Number: 1

Key	Type	Description	Command
Home	Menu	Intro	Change
Add: Menu Transfer Mailbox Information Park&Page			
1	Transfer	To 5000	Change Delete
2	Transfer	To 5001	Change Delete
3	Transfer	To ICC 1	Change Delete
4	Transfer	To ICC 2	Change Delete
5	Transfer	To ICC 3	Change Delete

Figure 22: CCR Tree 1 Menus

Greeting Table 1 Setup

	Morning	Afternoon	Evening	Non Business
Greeting:	1 ▼	1 ▼	1 ▼	1 ▼
CCR Tree:	1 ▼	1 ▼	1 ▼	1 ▼
Disable DN Dialing:	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Monday:	12:00 AM	12:00 PM	6:00 PM	6:00 PM (eg 12:00 AM)
Tuesday:	12:00 AM	12:00 PM	6:00 PM	6:00 PM (eg 12:00 AM)
Wednesday:	12:00 AM	12:00 PM	6:00 PM	6:00 PM (eg 12:00 AM)
Thursday:	12:00 AM	12:00 PM	6:00 PM	6:00 PM (eg 12:00 AM)
Friday:	12:00 AM	12:00 PM	6:00 PM	6:00 PM (eg 12:00 AM)
Saturday:	12:00 AM	12:00 PM	6:00 PM	6:00 PM (eg 12:00 AM)
Sunday:	12:00 AM	12:00 PM	6:00 PM	6:00 PM (eg 12:00 AM)
Attendant Extension:	<input type="text"/>			
Language Preference:	Primary ▼			
Menu Repeat Key:	None ▼			
Custom Auto-Attendant Menu Prompts				
Enable:	<input type="checkbox"/>			
Primary Prompt:	Not Recorded		<input data-bbox="984 1350 1128 1392" type="button" value="Voice..."/>	
Alternate Prompt:	Not Recorded		<input data-bbox="984 1409 1128 1451" type="button" value="Voice..."/>	

Figure 23: Auto-Attendant Greeting Table using CCR Tree 1

For more information on configuring Auto-Attendant and CCR Trees, refer to the *CallPilot Manager Set Up and Operation Guide* (NN40090-300).

Configuring CallPilot to Answer Inbound IP Toll-Free Calls

CallPilot Manager → Auto-Attendant → Lines Administration:

Click "**Change**" on the corresponding target lines configured in Section 5.1 to have CallPilot answer the line.

Set "**Answer Mode**" to "Auto-Attendant" to enable the Auto-Attendant to answer the call; select "Contact Center" to enable the Contact Center to answer the call directly.

Set "**Table/Skillset Number**" to Auto-Attendant Greeting Table number if Auto-Attendant is used; Skillset number if Contact Center is used.

Line Properties

Line Number:	998
Answer Mode:	Contact Center
Table/Skillset Number:	1
Number of Rings:	0

Figure 24: Line Properties in CallPilot Manager

Configuring Sets/Hunt Groups to Answer Inbound IP Toll-Free Calls

Similarly, if Intelligent Contact Center is not a requirement, inbound IP Toll-Free calls can be mapped directly to sets and/or hunt groups. Map the target lines (in Section 5.1) to the desired DNs in Element Manager.

Troubleshooting

This section provides some tips about troubleshooting problems.

System Monitoring with BCM Monitor

A valuable application for performance monitoring is the BCM Monitor. It allows the BCM administrator to see the current status of various parts of the BCM system. Statistical information is provided on system throughput and other performance-related information, including system CPU usage (graph or table format) and memory usage (graph or table format).

If a performance display is active, it is automatically updated with real-time performance information in user-selectable time increments.

The focus of the real-time monitoring capabilities is:

- Overall system status
- Utilization of resources on the Media Services Card (e.g. signaling channel usage)
- Operation of telephony applications (e.g., Messaging, Call Center, etc.).
- IP telephony activity
- D-channel monitoring for PRI, BRI and VoIP trunks

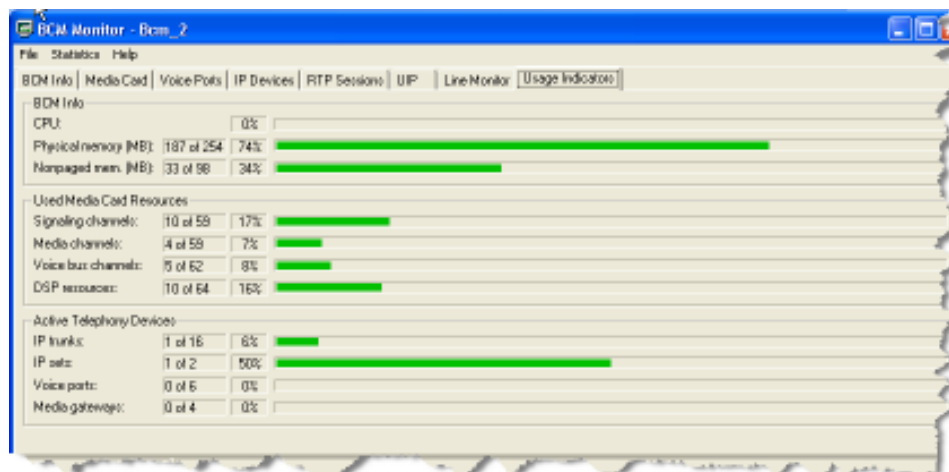


Figure 25: System Monitoring Example

The BCM Monitor application can be downloaded to an administrator's PC from the BCM and pointed at a specific BCM's IP address for monitoring. Multiple instances of the BCM

Monitor application can be used on a single PC to monitor several remote BCM systems at the same time.

Backward version compatibility is supported.

All of the registered IP devices can be viewed with the BCM Monitor. The screen shot below depicts IP Phone type, DN number and IP address of each registered IP phone. Additionally, if the device is active on a call the RTP session information is also displayed.

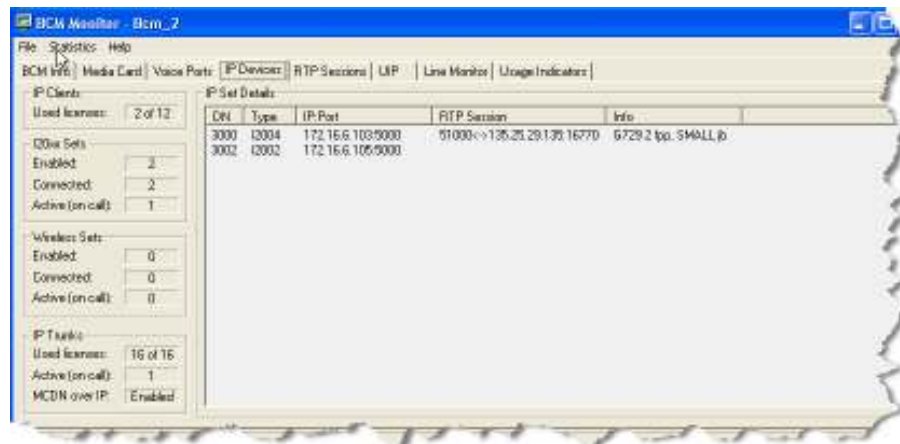


Figure 26: IP Device Listing

The end-to-end RTP sessions per IP call can also be displayed with the BCM Monitor. The example below depicts an end-to-end call.

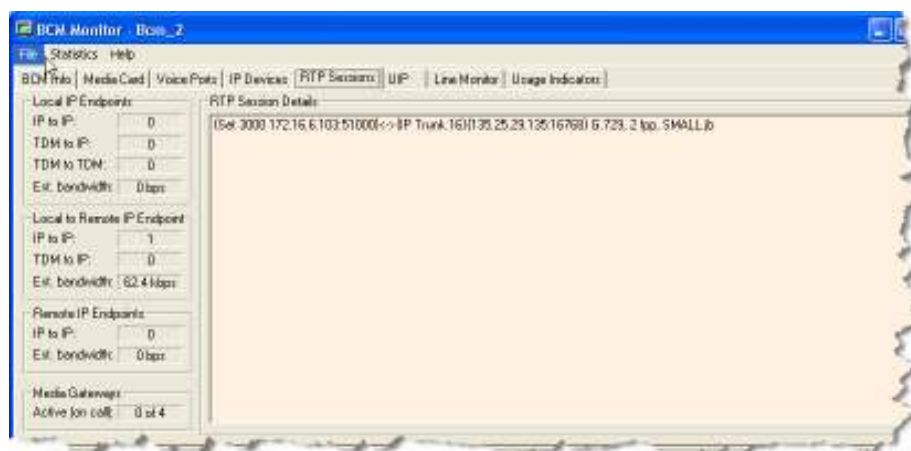


Figure 27: RTP Session Information

The BCM Monitor can be used to monitor incoming and outgoing trunks to determine if trunks are being busy or if they are idle. The example below depicts utilized lines used by local and remote telephone/DN numbers.

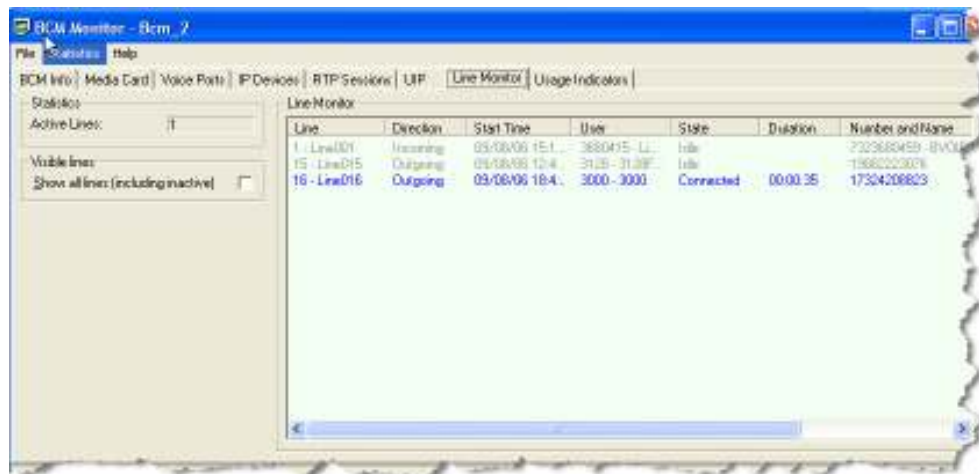


Figure 28: Line Monitor Information

The BCM Monitor can also be used to monitor all types of system usages. The following are some parameters that can be monitored:

- CPU utilization
- Physical memory
- Media card DSP utilization
- IP sets and IP Trunks
- Voice ports and media gateway usage

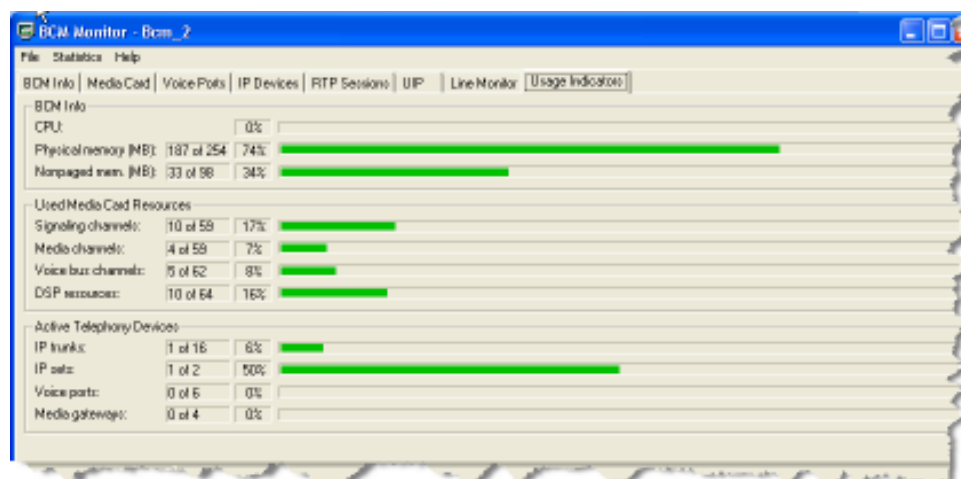


Figure 29: System Resources

Real-time display of BCM Alarms

Administration → General → Alarms:

The BCM provides extensive alarm logs along with severity and problem descriptions. The following is an example screen shot of the "Alarms" display:

The screenshot displays the 'Alarms' section of a BCM interface. It features a table with columns for Time, Alarm Acked, Alarm ID, Severity, Problem Description, and Component ID. The table lists several alarms, with the most recent one (2009-11-08 20:21) highlighted in blue. Below the table are buttons for 'Clear Alarm Log' and 'Reset Status LED'. A section titled 'Alarm Details' shows the time as 'Sun Nov 08 20:21:04 EST 2009'. It includes a 'Problem description' field with the text 'FindMe/FollowMe. Configuration of Virtual Terminal 5153 is wrong' and a 'Problem resolution' field with the text 'Verify that the Virtual Terminal has 3 intercom keys.' An 'Acknowledge Alarm' button is located at the bottom of the details section.

Time	Alarm Acked	Alarm ID	Severity	Problem Description	Component ID
2009-11-09 10:54	<input type="checkbox"/>	30200	information	User login User=mnadmin Host=10.10.11.2:1047 Comp=CIM	systemId=BCM;entityId=B...
2009-11-08 20:23	<input type="checkbox"/>	46806	information	Ran IVR advanced command <amu +t>	systemId=BCM;entityId=B...
2009-11-08 20:22	<input type="checkbox"/>	30200	information	User login User=mnadmin Host=172.16.10.249:3720 Comp=CIM	systemId=BCM;entityId=B...
2009-11-08 20:21	<input type="checkbox"/>	10909	information	System Startup - Startup complete. Service Manager and Schedu...	systemId=BCM;entityId=B...
2009-11-08 20:21	<input type="checkbox"/>	47006	minor	FindMe/FollowMe. Configuration of Virtual Terminal 5153 is wrong	systemId=BCM;entityId=B...
2009-11-08 20:21	<input type="checkbox"/>	47006	minor	FindMe/FollowMe. Configuration of Virtual Terminal 5154 is wrong	systemId=BCM;entityId=B...
2009-11-08 20:21	<input type="checkbox"/>	44000	information	Voicemail is operational	systemId=BCM;entityId=B...
2009-11-08 20:20	<input type="checkbox"/>	47009	information	FindMe/FollowMe. Registered Handoff Feature Code F960	systemId=BCM;entityId=B...
2009-11-08 20:20	<input type="checkbox"/>	8023	information	MCC Modem Enabled	systemId=BCM;entityId=B...

Clear Alarm Log Reset Status LED

Alarm Details. Time: Sun Nov 08 20:21:04 EST 2009

Problem description: FindMe/FollowMe. Configuration of Virtual Terminal 5153 is wrong

Problem resolution: Verify that the Virtual Terminal has 3 intercom keys.

Acknowledge Alarm

Figure 30: BCM Alarms Page

Log Management

Another useful tool is the “Log Management.” This allows you to quickly and easily collect all relevant logs files and other information to help the various support teams debug any problems you may have with your BCM. All required log files are consolidated into a single file.

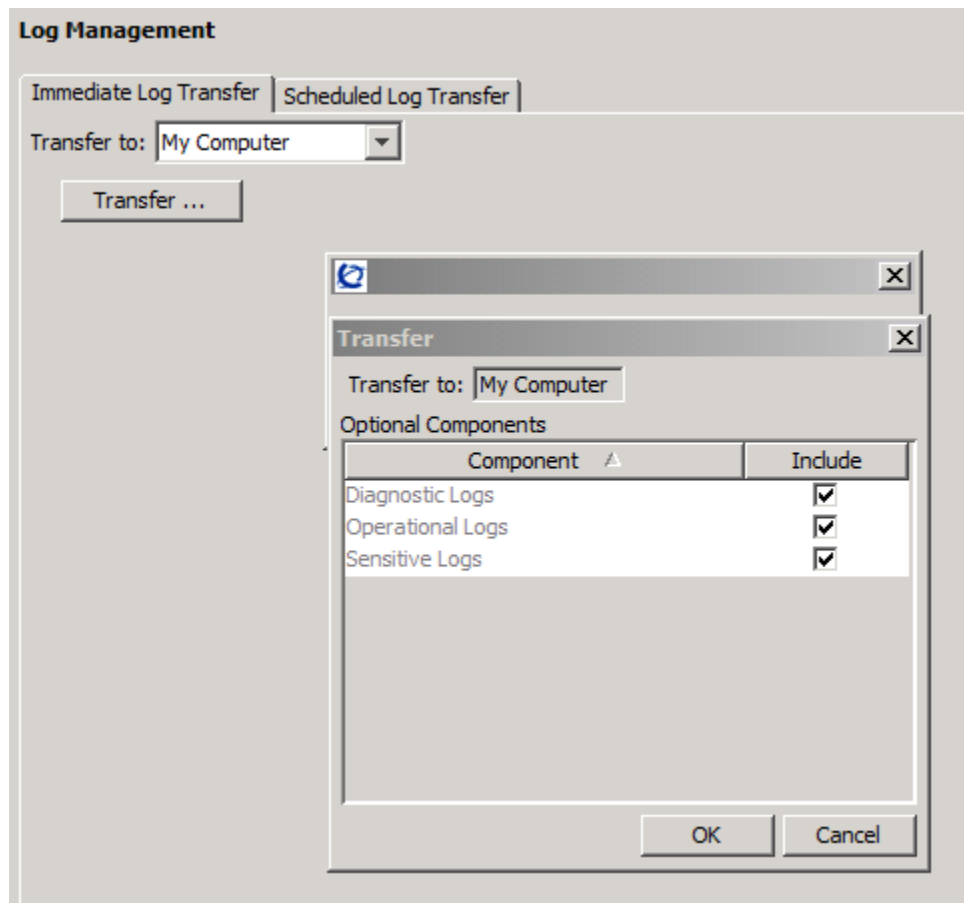


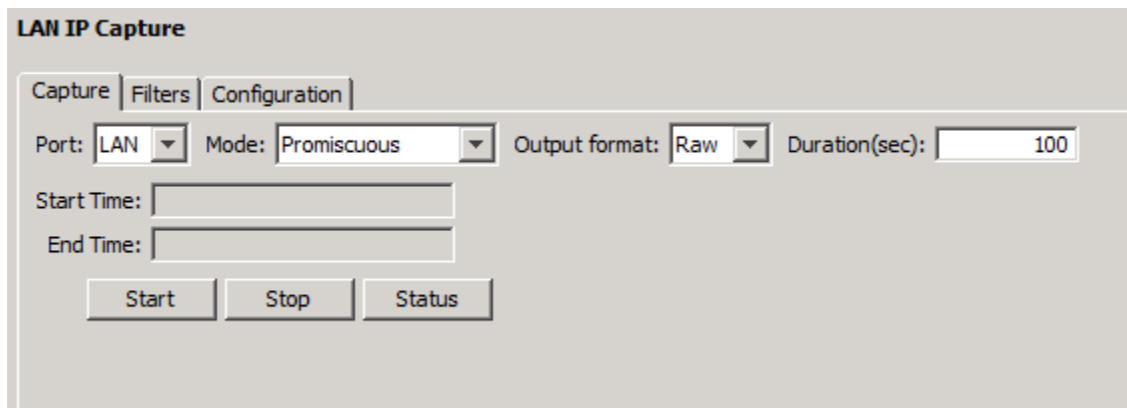
Figure 31: Log Management

When you first suspect a problem with your BCM, it is important that you go into the “Log Management” screen and download the log file to your PC. Even if you end up resolving the issue, it is good to know that this information has been captured if it does end up being required.

LAN Packet IP Capture Tool

BCM Release 5.0 includes a LAN packet capture tool, which can be initiated using the Element Manager. This tool will capture IP packets to/from the BCM, including SIP signaling, and store them into an output file on the BCM file system, a USB drive, or network drive. This is useful for troubleshooting issues such as SIP calls.

The tool can be found under ***Administration → Utilities → LAN IP Capture***.



The screenshot shows the 'LAN IP Capture' window with three tabs: 'Capture', 'Filters', and 'Configuration'. The 'Capture' tab is active. It contains the following fields and controls:

- Port:** A dropdown menu set to 'LAN'.
- Mode:** A dropdown menu set to 'Promiscuous'.
- Output format:** A dropdown menu set to 'Raw'.
- Duration(sec):** A text input field containing '100'.
- Start Time:** An empty text input field.
- End Time:** An empty text input field.
- Buttons:** Three buttons labeled 'Start', 'Stop', and 'Status' are located at the bottom.

Figure 32: LAN IP Capture Tool

Note: Depending on the network architecture, not all IP packets will be captured by this tool, i.e. RTP packets from the IP sets.

APPENDIX A: Configuring Destination Code with Wildcard

In an inbound call scenario, the leading digit sent to the BCM may be the same as the digit used in the destination code. Without configuring for a wildcard; the BCM will interpret the call as a tandem call, and will fail to terminate the call on the BCM. To remedy this, it is recommended to configure destination code with wildcard.

If the IP Flexible Reach sends a site prefix that is the same as the digit being used for the destination code; please use the following configuration example.

In this example, the number sent to the BCM is the following: "961170". To configure the BCM for this call, use the following wildcard configuration.

Configuration → Telephony → Dialing Plan → Routing:

Select ***Destination Codes*** tab, add the destination code "9A" to use Normal Route "001." Configure the absorbed length to 1 so that the BCM will absorb the '9' only in an outbound call scenario. Uncheck the digit following the '9' in the incoming digits to the BCM (in this example, the '6' in "961170").

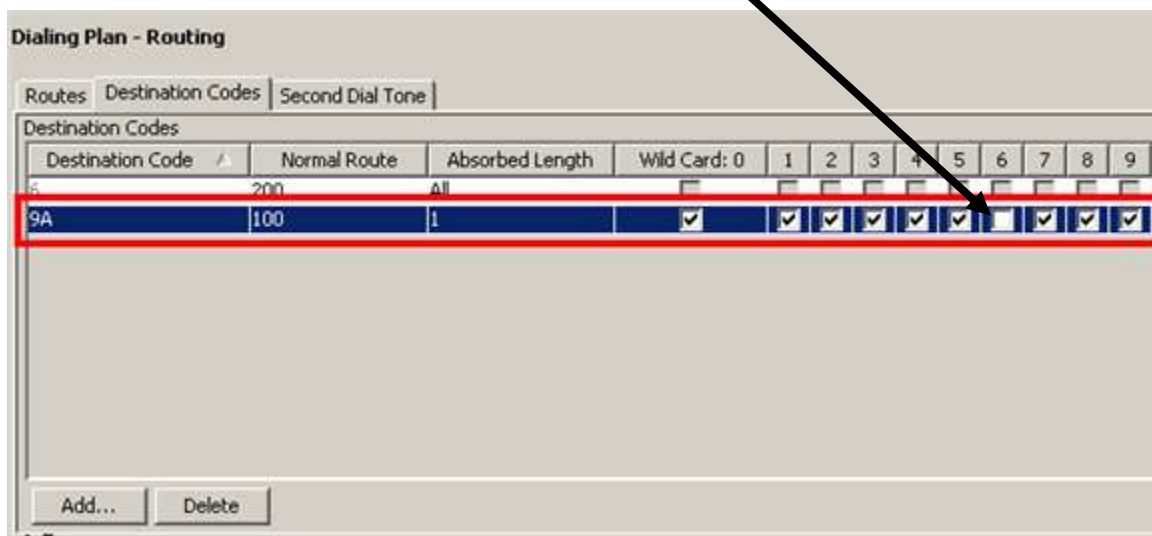


Figure 33: Configuring Destination Code with Wildcard

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